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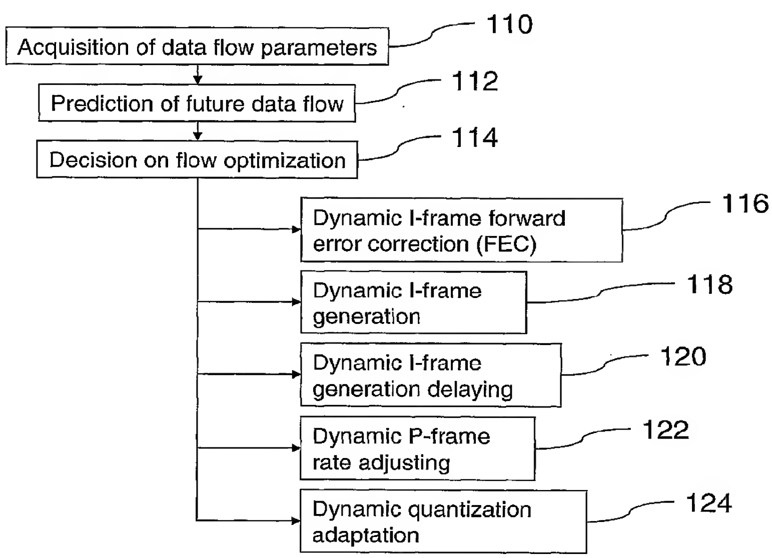
(54) Title: DYNAMIC OPTIMIZATION OF WIRELESS REAL-TIME VIDEO DATA FLOW



(57) Abstract: Quality of service management for video streaming applications is an important issue for wireless mobile devices, such as for upload or download of video data between personal digital assistants (PDAs) or 3G cellular phones, and a wireless communication network. The invention discloses a method for dynamically adapting a video stream during the transmission. The adaptation is mainly based on a prediction of the future state of the data flow. According to the predictions, a decider module decides on a method (116, 118, 120, 122, 124) for either adapting the encoding of the video data or for adapting the transmission of these data, depending on the kind of frame to be transmitted. By using this quality of service management, the mobile system can react to changes in the quality of the wireless link at an early stage and, thus, greatly improve the quality video data transmission.

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Dynamic optimization of wireless real-time video data flow

Field of the invention

The present invention relates to the quality of service (QoS) management for mobile video applications. More specifically, the invention relates to dynamic optimization of wireless real-time video data flow, based on general predictions of mobile link characteristics.

Background of the invention

By now, the standardization of 3rd-generation wireless systems is almost completed in most major economic regions of the world. These systems are known under names such as IMT-2000 (ITU), UMTS (ETSI 3GPP), EDGE and ANSI 3GPP2. A large number of suitable 3G wireless devices, such as cell phones or personal digital assistants (PDAs) are commercially available. The transfer of data using the above-mentioned systems is predominantly packed-switched, mostly using the well-known internet protocol (IP). The transport of IP packets over the air interface not only extends the reach of the internet, it also

opens the opportunity to migrate all of the communication to a packet switched environment.

Using IP protocol for mobile radio networks has its challenges, which are mainly due to frequent changes in the quality of the connection between the mobile user and the corresponding base stations. These changes are a result of a number of complex factors, such as geographical factors, meteorologic factors or the movement of the mobile user, resulting in frequent network cell changes.

The impact of these frequent changes in the quality of connection, resulting, e.g., in a frequent change of the bit error rate (BER) or the frame loss rate (FLR), on typical real-time applications (mostly including voice and/or video packet transfer) strongly depends on the application itself: In Email data transfer, e.g., the reliability of the packet transfer is an esential factor, whereas, e.g., the speed of the transfer is of minor importance. For real-time audio- and video applications on the other hand, the delay of the data packages has to be minimized, since the mobile user regards delayed sequences as highly disturbing, whereas missing packages (resulting, e.g., in "crackling" voice transfer) are less noticeable.

In PCT/EP 02/03018 and DE 102 47 581, quality of service state predictors are described, which use a method for predicting link quality parameters in 2.5 and 3G mobile access networks. The predicted link quality parameters are used for controlling lower layer corrective mechanisms such as transmission power control, to aid QoS systems and applications in their quality management process. E.g., the codec mode and the bit rate can be adapted according to the current and predicted link quality.

Nevertheless, since video encoding is largely different from the encoding schemes of voice packages, the method for adapting the data transfer described in PCT/EP 02/03018 or DE 102 47 581 is rather limited with regard to video streaming.

Modern video encoders, called codecs, such as the H.264 video codec, include features for adapting a video stream for a wide range of different links. Note that the terms "video" as well as "video streaming", in this context, are used in a technical meaning. They do not include audio and audio transmission but only the (moving) pictures.

The features for adapting a video stream also include features for adapting to links with low bandwidth and/or high bit error rate. In prior art solutions, this adaptation is static during the video session, i. e., once the parameters are fixed for a transmission session, they are not to be changed again until the transmission is over. This means that the video stream will be adapted to the underlying link only at the beginning of the video transmission.

Summary of the invention

It is therefore an object of the present invention to provide a method and a system for dynamic optimization of wireless real-time video data flow.

The invention is based on the finding that quality of the video transmission can be improved dramatically if the adaptation is done dynamically during the transmission knowing the actual status of the underlying link (e.g., bandwidth, bandwidth variations, delay, jitter, bit error rate) or even predict the (near) future development of these parameters.

Other objects and advantages of the present invention may be ascertained from a reading of the specification and appended claims in conjunction with the drawings.

Part of the invention is a method for dynamically adapting a video stream during the transmission. The preferred embodiments of the invention are set forth in the dependent claims.

A method for dynamic optimization of real-time video data flow between a mobile device, such as a personal digital assistant (PDA) or a 3G cellular phone, and a wireless communication network is diclosed. The mobile device is assumed to comprise at least one application generating and encoding video data using at least one codec. Further, a hardware system is disclosed, enabling the realization of the method in one of the listed variations.

As it is a standard for video data encoding, the encoded video data is assumed to be comprising P-frames and I-frames. I-frames are including the full (compressed) video picture. For a device receiving and decoding video data, in order to decode an I-frame, the decoder does not need information on previously send frames. The picture used for encoding this I-frame is the starting point for P-frame generation. P-frames are including information on how parts of previous I-frames are changing (e.g., shifts, size variations, etc.). This information is mainly vector based. Without the previously generated I-frame it is not possible to decode a video picture only by using a P-frame.

Generally, the amount of data of a P-frame is considerably smaller than the amount of data of an I-frame. Consequently, the transmission of P-frames requires a lower bandwidth than the transmission of I-frames.

In more or less regular temporal intervals, I-frames are sent, whereas in between, only P-frames are transmitted, in order to reduce the overall amount of transmitted data. Thus, e.g., every 200 P-frames, one I-frame is generated and transmitted.

The loss of a P-frame results in the loss of one decoded video picture, thus reducing the decoded video frame rate and dete-riorating the quality of the video. The loss of an I-frame re-sults in the loss of a whole sequence of video pictures.

The method described in the following comprises several steps. These steps not necessarily have to be taken in the given or-der. One or more steps can be performed in parallel. Addi-tional steps not listed can be performed.

First, a set of actual parameters indicating the current state of the data flow are acquired. These parameters preferrably comprise one or more of the following parameters:

- parameters of the air link, as e.g. coding scheme, and/or
- parameters of a transmission protocol stack, and/or
- available bandwidth, and/or
- maximum buffer sizes, and/or
- buffer fill levels, and/or
- information about PDP (Packet Data Protocol) contexts, e.g. quality of service (QoS) settings for the PDP context, and/or
- radio resource management information, and/or
- received signal code power (RSCP), and/or
- signal to interference ratio (SIR), and/or
- received signal strength indicator (RSSI), and/or
- signal strength of the wireless connection, and/or
- traffic volume measurement, and/or
- position of the mobile device, and/or
- altitude of the mobile device, and/or

- direction of the mobile device, and/or
- velocity of the mobile device, and/or
- block size (i.e. the size of the Data Link Layer transmission blocks. Remember that IP packets are cut in blocks and the blocks are transferred to the receiving side.), and/or
- block error rate (which is similar to the bit error rate but seen for the whole Data Link Layer transmission block), and/or
- the codec employed, and/or
- the compression of header data, and/or
- bit error rate, and/or
- frame loss rate, and/or
- transmission delay.

In a next step, on the basis of these actual parameters and other necessary data, a prediction of a future state of the data flow is made for a given time-interval. As an example, this prediction may refer to one or more of the following types of information on the data flow:

- predictions related to cell reselections, and/or
- predictions related to throughput, and/or
- predictions related to signal to interference ratio (SIR), and/or
- predictions related to bit error rate, and/or
- predictions related to the suitable coding scheme (In GPRS and EDGE (at least) a certain number of bits, i.e. a transmission block can transferred in one timeslot. Some of the bits can be used for the data transfer, others are used to secure these data bits. There are existing different coding schemes (4 in GPRS, 9 in EDGE) which are standing for a different ratio of data bits / protection bits. This means: using coding scheme one you have the highest protection for the data bits but the lowest number of usable bits for

data transfer, coding scheme 4 provides less protection but more bits for data transfer, resulting in a higher bandwidth.), and/or

- predictions related to transmission delay, and/or
- predictions related to block error rate (an error in a transmission block (if the above mentioned protection failed) are leading to an error in the IP packet which may lead to the loss of an IP packet. Predicting this rate means predicting packet losses or damaged packets), and/or
- predictions related to round trip time, and/or
- predictions related to the increased and decreased bandwidth available for transmission of the video data.

Algorithms for calculating predictions of this type are state of the art and are disclosed, e.g., in PCT/EP 02/03018 or DE 102 47 581. The meaning of the expression "time-interval" is not necessarily restricted to an actual time, it can, e.g., equally well designate an internal clock of a computer. Other time scales, not necessarily having a continuous and steady succession in time, but indicating, e.g., the progress of a transmission, might be used. A widely used time scale is the TTI-time-scale (transmission time interval, i.e. 10 or 20 ms per interval).

In a next step, one or more measures are taken, in order to dynamically adapt the video data flow, especially in order to reduce the risk of loss of important video data, to the predicted state of the data flow during the given time-interval in the near future. The goal of this adaptation is to provide the best video quality on the receiving side for each situation, i. e. for each possible state of the the link or data flow quality. The measures may comprise one or more of the following steps:

- dynamic I-frame forward error correction (FEC)
- dynamic I-frame generation
- dynamic I-frame generation delaying
- dynamic P-frame rate adjusting
- dynamic quantization adaptation

These steps are described in detail in the following. Some of the steps may be performed in one or more of the described variations.

Dynamic I-frame forward error correction (FEC)

In a first preferred embodiment, the method of dynamic I-frame forward error correction (FEC) is employed. Dynamic I-frame FEC is typically used in cases of high bit error rates with the goal to prevent the loss of an I-frame, which would result in the loss of a whole sequence of frames in a row on the de-coded side. Note that P-frames based on a lost I-frame cannot be de-coded.

Two possible cases may be considered for this method:

1. The loss of an I-frame during transmission is detected.
2. It can be predicted (or estimated) that the probability to lose a certain I-frame during transmission is high.

Especially (but not solely) in the first case, a copy of each recently sent I-frame is buffered, and if a loss of this I-frame during transmission is detected, the I-frame is re-transmitted. There is no direct acknowledgment for receiving an I-frame. But the RTCP packets (RTP Control Protocol or Real Time Control Protocol), used for quality feedback information for the RTP streams, can be used to give a feedback from the receiver to the sender.

In a preferred embodiment, the I-frame is re-transmitted with additional forward error correction (FEC) information. This step may be combined with the additional condition that the time that has passed between the generation of the respective I-frame and the detection of the loss of this I-frame is not too high, i. e. remains below a given threshold.

Several forward error correction means are known to the person skilled in the art and may be used. These forward error correction algorithms include adding additional data allowing for the detection of transmission failures and the reconstruction of certain data packages by the receiving device, even if part of the transmitted data are lost during transmission. Besides forward error correction, other means of error correction usually employed in wireless data transfer can be employed accordingly. These are, e.g., convolutional coding or bit coding according to the used coding scheme (see above).

Especially (but not solely) in the second case listed above, the following method is proposed: If the prediction of the state of the data flow for a given time-interval in the future indicates a risk of loss of the frames to be transmitted within this time-interval being above a pre-defined risk level, the I-frames to be transmitted within this time-interval are send with additional FEC information. This additional FEC information increases the chance that (even if part of the transmitted data are lost), the I-frame may be reconstructed by the addressee.

In a preferred embodiment, when using the method of dynamic I-frame forward error correction, the generation of P-frames as well as the transmission of these P-frames remains independent of the predicted future state of the data flow.

Dynamic I-frame generation

In a second preferred embodiment, the method of dynamic I-frame generation is used with the goal to send a new I-frame, i.e. synchronization point, to the decoder of the frames in the receiving device.

Dynamic I-frame generation might especially be useful or even necessary in one of the following cases:

1. The loss of an I-frame could not be prevented and there was no possibility to re-transmit a copy of the lost I-frame.
2. The last I-frame was generated more than a given time-interval ago. In this case the video pictures are de-coded on the basis of the P-frames combined with a rather "old" I-frame. This will reduce the accuracy of the P-frames so that the difference between encoded and decoded picture are increasing. One reason for this situation may be a delayed I-frame generation as a con-sequence of a reduced available bandwidth (dynamic I-frame generation delaying, see below). If more band-width becomes available, the generation of an I-frame may be enforced.

Thus, the following step is proposed: If since the last sucessful transmission of an I-frame a time-period longer than a pre-defined time-period has passed, one or more control sig-nals are generated. These control signals trigger a codec to create an I-frame at the nearest possible point in time.

In a preferred embodiment, when using the method of dynamic I-frame generation, the generation of P-frames as well as the transmission of P-frames remains independent of the predicted future state of the data flow.

Dynamic I-frame generation delaying

In a third preferred embodiment, dynamic I-frame generation delaying is used. This method is especially useful in cases where the bandwidth available for data transmission is reduced.

Thus, if the prediction of the state of the data flow for a given time-interval in the future indicates a risk of loss of the frames to be transmitted within this time-interval being above a pre-defined risk level, the following means may be taken: First, one or more control signals may be generated controlling the at least one codec not to create an I-frame within this time-interval. Alternatively or additionally, I-frames to be sent within this time-interval are buffered, and the transmission of these I-frames is delayed until the time-interval has passed.

During this period of delay, only P-frames are transmitted. As described above, the transmission of P-frames typically requires a lower bandwidth. Nevertheless, the vector based information encoded in the P-frames are becoming more inaccurate with the rising "age" of the corresponding I-frame resulting in an inaccuracy of the decoded video picture.

In a preferred embodiment, when using the method of dynamic I-frame generation delaying, the generation of P-frames as well as the transmission of P-frames remains independent of the predicted future state of the data flow.

Dynamic P-frame rate adjustment

In a fourth preferred embodiment, dynamic P-frame rate adjustment is used, preferrably in those cases where the bandwidth available for data transfer is too low for a transfer of all P-frames.

Thus, if the prediction of the state of the data flow for a given time-interval in the future indicates a limitation of the amount of data that can be transmitted at a given quality below a given level, the transmission of one or more P-frames to be sent within this time-interval may be suppressed. In this case single P-frames are dropped (i. e. erased from a transmission buffer) rather than transmitted, in order to reduce the number of P-frames. Preferrably, P-frames are not dropped "in sequence" but out of sequence, stochastically, in order to prevent "jumps" in the decoded video stream.

Alternatively, instead of simply suppressing the transmission of these P-frames, one or more control signals may be generated controlling the at least one codec not to create a P-frame within this time-interval or to reduce the number of P-frames to be generated within this time-interval below a certain number or rate.

Dynamic quantization adaptation

In a fifth preferred embodiment, dynamic quantization adaptation is used to adapt the amount of transferred data according to the available bandwidth. Note that there is a negative relationship between the quantization and the quality of a transmitted video picture received and decoded by an addressee: Increasing the quantization results in a reduced quality of the decoded video picture and vice versa. Quantization is data reduction.

Quantization typically is the "lossy" part of the video picture compression. It can be compared with the JPEG compression of a (still) picture, as known to the person skilled in the art. With higher quantization the quality of the decoded picture is lower but also the needed bandwidth to transfer this video picture is reduced. The video codec H.264, e.g., contains 52 quantization levels.

Thus, if the prediction for the state of the data flow for a given time-interval in the future indicates that the amount of data that can be transmitted will be above a given level (i. e. in the case of high available bandwidth), one or more control signals may be generated controlling the at least one codec to reduce the quantization of the data, thus increasing the quality of the video data. On the other hand, if the predictions indicate a low available bandwidth, the quantization of the data may be increased.

Note that whereas the first four methods and embodiments are working on a (very granular) frame level, quantization adaptation will have an influence on a whole group of frames according to long-time but bigger bandwidth changes (on a temporal timescale of app. 500 ms). The other four methods typically might be of special use in cases of occurrence of very fast but rather small short-time changes of the state of the data flow.

Also note that, if besides the video stream an audio stream is to be transmitted, the audio stream typically will have the higher priority. As described above, the mobile device user typically is willing to accept some lost video pictures rather than lost voice fragments. Thus, in this document, the term "available bandwidth" for video streaming is defined as:

available bandwidth = provided bandwidth - audio bandwidth.

Furthermore, the present invention includes:

- a computer loadable data structure that is adapted to perform the method according to one of the embodiments described in this description while the data structure is being executed on a computer,

- a computer program, wherein the computer program is adapted to perform the method according to one of the embodiments described in this description while the program is being executed on a computer,

- a computer program comprising program means for performing the method according to one of the embodiments described in this description while the computer program is being executed on a computer or on a computer network,

- a computer program comprising such program means, wherein the program means are stored on a storage medium readable to a computer,

- a storage medium, wherein a data structure is stored on the storage medium and wherein the data structure is adapted to perform the method according to one of the embodiments described in this description after having been loaded into a main and/or working storage of a computer or of a computer network, and

- a computer program product having program code means, wherein the program code means can be stored or are stored on a storage medium, for performing the method according to one of the embodiments described in this description, if the pro-

gram code means are executed on a computer or on a computer network.

Brief description of the drawings

For a more complete understanding of the present invention, reference is made to the following description made in connection with accompanying drawings in which:

Fig. 1 shows a a schematic overview of the method for dynamic optimization of real-time video data flow between a mobile device and a wireless communication network; and

Fig. 2 shows a schematic diagram of a system for performing the method depicted in Fig. 1 in one of its embodiments;

Detailed description of preferred embodiments

In Fig. 1, a schematic overview of the method for dynamic optimization of real-time video data flow between mobile devices and a mobile network is depicted. In Fig. 2, a physical and/or embedded system is depicted, adapted for realizing the method of Fig. 1 in one or more of its variations. The arrows in Fig. 2 indicate the direction of data flow. In the following, Fig. 2 will be described in conjunction with the respective steps depicted in Fig. 1.

On the left hand side, Fig. 2 shows the typical system of functional layers of a mobile device real-time video streaming as known to the person skilled in the art from the OSI reference model. Out of the seven OSI layers, in Fig. 2 only the Application Layer 210, the Transport Layer 212, the Data Link

Layer 214, and the Physical Layer 216 are depicted. The Network Layer, the Session Layer and the Presentation Layer are omitted for the sake of simplicity, but may be controlled in a similar way.

On the level of the Application Layer 210, several applications 218 comprising applications generating video data may be run. As an example, video data acquisistion using a cell phone equipped with a video camera may be named, including the respective application software. Also included in the Application layer are one or more codec modules 220, 222, including codecs for video encoding 220 and for voice encoding 222. As an example, the implementation of the video codec H.264 and the audio codec AMR (adaptive multi rate) is assumed in the following. The codecs 220, 222 transform the data streams generated by the various applications 218 into encoded data frames 224, which are passed down from the Application Layer 210 via the various other layers to the Physical Layer 216 to be transmitted via the wireless network.

On their way down to the Physical Layer 216, in each of the various OSI layers, the frames 224 may be modified (symbolically depicted by the frames 226 in Fig. 2), especially equipped with additional information (e.g., additional headers), according to the respective protocols used for the type of information to be transmitted. Thus, in the example depicted in Fig. 2, in the Transport Layer 212, the Real-Time Transport Protocol (RTP) 228, the User Datagram Protocol (UDP) 230, and the Internel Protocol (IP) 232 are employed. In the Data Link Layer 214, the Logical Link Control Layer (LLC) 248, the Radio Link Control Layer (RLC) 250, and the Medium Access Control Layer (MAC) 251 are employed in this embodiment.

Each layer is equipped with one or more control modules 234 – 246. These control modules control the functionality of the

layers in various ways, depending on the layer itself. Thus, e.g., the Media Control Module 234 controls the settings of the audio and/or video codec(s) (e.g., the quantization, see above) of the application layer. The RTP control module 236 controls the FEC (forward error correction) packet generation, the I-frame buffering and retransmission as well as the reading out of RTCP (RTP Control Protocol or Real Time Control Protocol) quality feedback information.

In the example depicted in Fig. 2, in the Data Link Layer 214, the Logical Link Control Layer (LLC) 248 and the Radio Link Control Layer (RLC) 250 are controlled by a common control module 242 (RRC, Radio Ressource Control).

Further, most or all of the layers have one or more buffers 252, 254 (symbolically depicted by the hatched boxes in Fig. 2) at their disposal. These buffers may be used for different purposes, such as for storing I-frames for delayed transmission when the prediction of the future state of the data flow indicates a high risk of loss (see above).

Besides controlling the settings and parameters of the single OSI layers, the control modules 234 - 246 allow for an easy access to actual parameters of the data flow. Thus, e.g., by accessing the control modules 242 and 244 of the Data Link Layer 214, Information on the quality of the transmission (e.g., the Bit Error Rate BER) can be gained. Further, information on the available ressources in each layer, e.g., the fill levels of the various buffers 252, 254, can be obtained. Acquiring these Actual Parameters 256 is the first step 110 of the method depicted in Fig. 1.

These Actual Parameters are passed on to a State Predictor Module 258. The State Predictor Module, based on the information 256, estimates the development of one or more relavant

variables indicating the state of the data transfer for the near future (step 112 in Fig. 1). In the preferred embodiment, for this purpose, the algorithm disclosed in DE 102 47 581is used. Thus, the State Predictor Module 258 may predict that the Bit Error Rate (BER) will be below a level of 10^{-9} for the upcoming 10 TTIs (transmission time intervals). These predictions 260 are passed on to a Decider Module 262.

The Decider Module 262 compares the predicted parameters 260 with a set of parameters stored in a Lookup-Table 264. In this Lookup-Table, which preferrably consists of a multi-dimensional matrix, the possible states of the future flow control predicted by the State Predictor 258 are divided into a numer of "cases", i. e. into a number of intervals for each relevant predicted parameter. Thus, for each case, a set of control parameters is referenced in this Lookup-Table.

Thus, in this embodiment, the "decision" on flow optimization the Decider Module 262 takes in step 114, basically, has the form of a certain set of control parameters 266, which are picked from the Lookup-Table 264 according to the Predictions 260 of the State Predictor 258. These Control Parameters 266 are passed on to the respective control modules 234 - 246, in order to adjust the data flow.

As disclosed above, several ways of controlling the data flow are possible (steps 116 - 124 in Fig. 1). The steps disclosed in this invention all refer to optimization of video streaming of encoded video data, but means for optimizing the data flow of encoded audio data may ba taken in parallel.

First, the decider may decide that according to the predictions of the future data flow, the method dynamic forward error correction (FEC) 116 may be employed. As explained above,

this method will preferrably be chosen when high BERs are predicted for the near future.

When dynamic I-frame forward error correction (FEC) 116 is chosen, control parameters for several control modules may be generated. First, control parameters controlling the Transport Layer 212 or the Data Link Layer 214 to store the most recently transmitted I-frame in one of the buffers 252, 254 may be generated and passed on to one of the control modules 236 - 244. In the preferred embodiment, the I-frame is buffered in the buffer 252 of the RTP 228. Thus, the I-frame can be retransmitted in case a loss during transmission occurs. Besides the RTP 228, there is a RTCP (RTP Control Protocol or Real Time Control Protocol) and an FEC for RTP module (both not shown). The RTCP will get the quality feedback packet from the receiving side. The decider 262 uses these pieces of information to decide that the I-frame should be retransmitted with additional FEC and signals to the RTP and FEC for RTP modules to retransmit the packet and to create an FEC packet.

In this context, when using the term "I-frames", it is obvious that not only the actual I-frames (224 in Fig. 2) are meant, but that the term also may include "modified" I-frames 226, i. e. after having passed layers below the Application Layer 210. These frames, as described above, also include additional information, such as additional headers.

Further, control parameters controlling the Physical Layer 216 or the Data Link Layer 214 to apply a certain schedule of error correction, especially forward error correction (FEC) may be passed on to the control modules 236 - 246. Thus, as an example, in case a BER above 10^{-3} is predicted for the upcoming 10 TTIs, the Decider Module 262 may control the Data Link Layer Control Module RRC 242 to increase a redundancy factor

(i.e. the factor controlling the error correction information) from 1.15 to 1.30 or change the coding scheme.

As explained above, the methods may be combined. E.g., the control parameters may first control the layers to buffer an I-frame and then, in case a loss of this I-frame is detected, to re-transmit it with increased redundancy factor.

Secondly, the Decider Module 262 may decide that the method of dynamic I-frame generation (118 in Fig. 1) is to be applied. As explained above, this method is especially useful in case the Actual Parameters 256 indicate that the loss of an I-frame could not be prevented and there was no possibility to re-transmit a copy of the lost I-frame or if since the last I-frame generation more than a pre-defined time-interval has passed. In case of dynamic I-frame generation 118, control parameters 266 for the Media Control Module 234 are generated controlling the codec 220 to create an I-frame at the nearest possible point in time.

Thirdly, the Decider Module 262 may decide that the method of dynamic I-frame generation delaying 120 is to be employed, which is especially useful if the Actual Parameters 256 indicate that the bandwidth available for data transmission is below a given level, an information which can be gained from a readout of the parameters of the Control Modules 242 - 246 by determining the used coding scheme in layer 1, the physical layer, the allocated timeslots in the data ling layer and the allocated transmission blocks in these timeslots, which can be collected from the RLC 250.

In this case, i.e. if the prediction indicates a high risk of loss of the frames to be transmitted in the near future, the decider may provide control parameters 266 to the Media Control Module 234 preventing the codec 220 from creating an I-

frame within a pre-defined time-interval. Alternatively or additionally, the decider may provide control parameters 266 to the control modules 234 - 246 of one of the layers, preferably of the Transport Layer 212 or of the Data Link Layer 214 to store the I-frames to be sent within this temporal interval in one of their buffers 252, 254 rather than to transmit them.

As soon as the temporal interval has passed, the fill-level of the buffers may be checked as part of the Actual Parameters 256, in order for the Decider 262 to make a new decision about transmission of the buffered data.

Fourthly, the Decider Module 262 may decide that the method of dynamic P-frame rate adjustment 122 is to be applied. As mentioned above, this might be the case especially when the Actual Parameters 256, in particular mainly control parameters of the Data Link Layer 214 or the Physical Layer 216 indicate that the bandwidth available for data transfer is too low for a transfer of all P-frames.

In this case, the Decider Module 262 may generate two types of control parameters 266: First, control parameters controlling the Application Layer 210, the Transport Layer 212, the Data Link Layer 214, the Physical Layer 216, or another layer not depicted in Fig. 2, to erase a certain number of P-frames upcoming for transmission from one or more of the buffers, e.g., from the buffers 252, 254. As indicated above, preferrably, these P-frames to be erased are selected stochastically rather than in sequence.

Further, the Decider Module 262 may generate control parameters for the Media Control Module 234 of the Application Layer 210 controlling the codec 220 not to create a P-frame within a given time-interval in the future (e.g., the upcoming 10 TTIs)

or to reduce the number of P-frames to be generated within this time-interval below a certain number or rate.

Fifthly, the Decider Module 262 may decide that the method of Dynamic quantization adaptation 124 is to be applied. As mentioned above, this method may be chosen in cases where the State Predictor 258 indicates that the state of the data flow for a given time-interval in the future, e.g., the upcoming 10 TTIs, will be such that the amount of data that can be transmitted will be above a given level. In this case, the Decider Module 262 may generate one or more control parameters 266 controlling the Media Control Module 234 to operate the codec 220 in a way that the quantization of the data for this time-interval is reduced, thus increasing the quality of the video data.

While the present inventions have been described and illustrated in conjunction with a number of specific embodiments, those skilled in the art will appreciate that variations and modifications may be made without departing from the principles of the inventions as herein illustrated, as described and claimed. The present inventions may be embodied in other specific forms without departing from their spirit or essential characteristics. The described embodiments are considered in all respects to be illustrative and not restrictive. The scope of the inventions are, therefore, indicated by the appended claims, rather than by the foregoing description. All changes which come within the meaning and range of equivalence of the claims are to be embraced within their scope.

Reference List

110	Acquisition of Actual Parameters of the data flow
112	Prediction of the future state of the data flow
114	Decision on flow optimization
116	Dynamic I-frame forward error correction
118	dynamic I-frame generation
120	dynamic I-frame generation delaying
122	dynamic P-frame adjusting
124	dynamic quantization adaptation
210	Application Layer
212	Transport Layer
214	Data Link Layer
216	Physical Layer
218	application
220	video codec
222	voice codec
224	data frames
226	modified data frames
228	Real Time Protocol, RTP
230	User Datagram Protocol, UDP
232	Internet Protocol, IP
234	Media Control Module
236	RTP Control Module
238	UDP Control Module
240	IP Control Module
242	Common Control Module RRC (Radio Ressource Control) for LLC (Logical Link Control Layer) and RLC (Radio Link Control Layer)
244	Control Module for MAC (Medium Access Control Layer)
246	Control Module for Physical Layer
248	Logical Link Control Layer (LLC)
250	Radio Link Control Layer (RLC)
251	Medium Access Control Layer (MAC)

252 buffer

254 buffer

256 Actual Parameters of the data flow

258 State Predictor Module

260 Predictions on the future state of the data flow

262 Decider Module

264 Lookup-Table

266 Set of Control Parameters

Claims

1. A method for dynamic optimization of real-time video data flow between a mobile device, such as a personal digital assistant (PDA) or a 3G cellular phone, and a wireless communication network;

- wherein the mobile device comprises at least one application generating and encoding video data using at least one codec;

- wherein the encoded video data comprise at least one full (compressed) video picture (I-frame) and at least one frame containing only the changes of a video picture since the last I-frame (P-frame);

- the method comprising the following steps:

a) actual parameters indicating the current state of the data flow are acquired;

b) on the basis of the actual parameters, parameters indicating a future state of the data flow are predicted for a given time-interval;

c) according to the predicted future state of the data flow, one or more of the following steps of dynamic flow optimization are taken:

c1) a copy of each recently sent I-frame is buffered, and if a loss of a recently sent I-frame during transmission is detected, the buffered I-frame is re-transmitted; and/or

c2) if the prediction of the state of the data flow for a given time-interval in the future indicates a risk of loss of the frames to be transmitted within this time-interval being above a pre-defined risk level, the I-frames to be transmitted within this time-interval are sent with additional FEC information; and/or

c3) if since the last sucessful transmission of an I-frame a time-period longer than a pre-defined time-period has passed, one or more control signals are generated controlling

the codec to create an I-frame at the nearest possible point in time; and/or

c4) if the prediction of the state of the data flow for a given time-interval in the future indicates a risk of loss of the frames to be transmitted within this time-interval being above a pre-defined risk level, one or more control signals are generated controlling the codec not to create an I-frame within this time-interval; and/or

c5) if the prediction of the state of the data flow for a given time-interval in the future indicates a risk of loss of the frames to be transmitted within this time-interval being above a pre-defined risk level, I-frames to be sent within this time-interval are buffered, and the transmission is delayed until the time-interval has passed; and/or

c6) if the prediction of the state of the data flow for a given time-interval in the future indicates a limitation of the amount of data that can be transmitted below a given level, the transmission of one or more P-frames to be sent within this time-interval is suppressed; and/or

c7) if the prediction of the state of the data flow for a given time-interval in the future indicates a limitation of the amount of data that can be transmitted below a given level, one or more control signals are generated controlling the codec not to create a P-frame within this time-interval or to reduce the number of P-frames to be generated within this time-interval; and/or

c8) if the prediction of the state of the data flow for a given time-interval in the future indicates that the amount of data that can be transmitted will be at a given level, at least one control signal is generated controlling the codec to adapt the quantization of the video data to be transmitted.

2. A method according to the previous claim,
characterized in that

- one or more of the steps c1), c2), c3), c4) or c5) are performed; and

- that the encoding and transmission of P-frames remains independent of the predicted future state of the data flow.

3. A method according to one of the previous claims, **characterized in that**

- step c6) is performed in a way that P-frames, which are selected to be suppressed during transmission, are chosen stochastically out of the video data stream.

4. A method according to one of the previous claims, **characterized in that**

the decision, which of the steps c1) - c7) is to be taken and/or the set of parameters to be employed for dynamic flow optimization, is based on a set of pre-defined decisions recorded in a lookup-table.

5. A method according to one of the previous claims, **characterized in that**

audio data are transmitted in parallel to the video data.

6. A method according to one of the previous claims, **characterized in that**

in step c1) the buffered I-frame is re-transmitted with additional forward error correction (FEC) information.

7. A system for dynamic optimization of real-time video data flow between a mobile device, such as a personal digital assistant (PDA) or a 3G cellular phone, and a wireless communication network;

- wherein the mobile device comprises at least one application generating and encoding video data using at least one codec;

- wherein the encoded video data comprise at least one full (compressed) video picture (I-frame) and at least one frame containing only the changes of a video picture since the last I-frame (P-frame);

the system comprising:

a) means for acquisition of actual parameters indicating the current state of the data flow;

b) means for predicting, based on the actual parameters, parameters indicating a future state of the data flow for a given time-interval;

c) means for choosing one or more of the following steps of dynamic flow optimization according to the predicted future state of the data flow:

c1) buffering a copy of each recently sent I-frame, and if a loss of a recently sent I-frame during transmission is detected, re-transmitting the buffered I-frame; and/or

c2) if the prediction of the state of the data flow for a given time-interval in the future indicates a risk of loss of the frames to be transmitted within this time-interval being above a pre-defined risk level, sending the I-frames to be transmitted within this time-interval with additional FEC information; and/or

c3) if since the last sucessful transmission of an I-frame a time-period longer than a pre-defined time-period has passed, generating one or more control signals, controlling the codec to create an I-frame at the nearest possible point in time; and/or

c4) if the prediction of the state of the data flow for a given time-interval in the future indicates a risk of loss of the frames to be transmitted within this time-interval being above a pre-defined risk level, generating one or more control signals, controlling the codec not to create an I-frame within this time-interval; and/or

c5) if the prediction of the state of the data flow for a given time-interval in the future indicates a risk of loss of

the frames to be transmitted within this time-interval being above a pre-defined risk level, buffering I-frames to be sent within this time-interval, and delaying the transmission until the time-interval has passed; and/or

c6) if the prediction of the state of the data flow for a given time-interval in the future indicates a limitation of the amount of data that can be transmitted below a given level, suppressing the transmission of one or more P-frames to be sent within this time-interval; and/or

c7) if the prediction of the state of the data flow for a given time-interval in the future indicates a limitation of the amount of data that can be transmitted below a given level, generating one or more control signals controlling the codec not to create a P-frame within this time-interval or to reduce the number of P-frames to be generated within this time-interval; and/or

c8) if the prediction of the state of the data flow for a given time-interval in the future indicates that the amount of data that can be transmitted will be at a given level, generating at least one control signal controlling the codec to adapt the quantization of the video data to be transmitted.

8. At least one of an operating system, a computer readable medium having stored thereon a plurality of computer-executable instructions, a co-processing device, a computing device and a modulated data signal carrying computer executable instructions for performing the method of one of the previous claims referring to a method.

9. At least one computer readable medium comprising computer executable modules including computer executable instructions for dynamic optimization of real-time video data flow between a mobile device, such as a personal digital assistant (PDA) or a 3G cellular phone, and a wireless communication network, wherein the mobile device comprises at least

one application generating and encoding video data using a codec, wherein the encoded video data comprise at least one full (compressed) video picture (I-frame) and at least one frame containing only the changes of video pictures (P-frame), the computer executable modules comprising means for performing one or more of the steps c1) - c8) in claim 1.

10. At least one of an operating system, a co-processing device, a computing device and a modulated data signal carrying the computer executable instructions of the computer executable modules of the at least one computer readable medium of the previous claim.

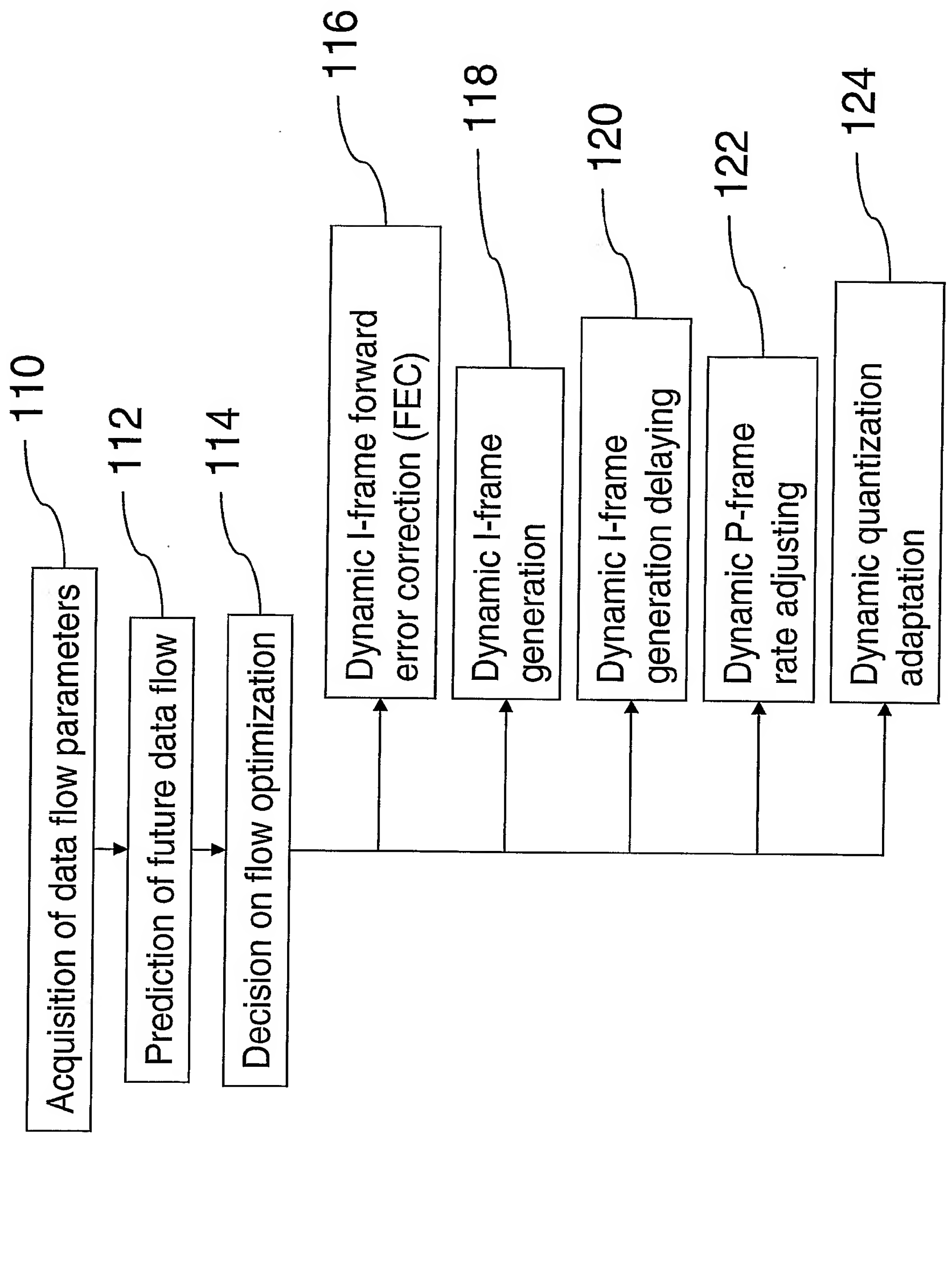
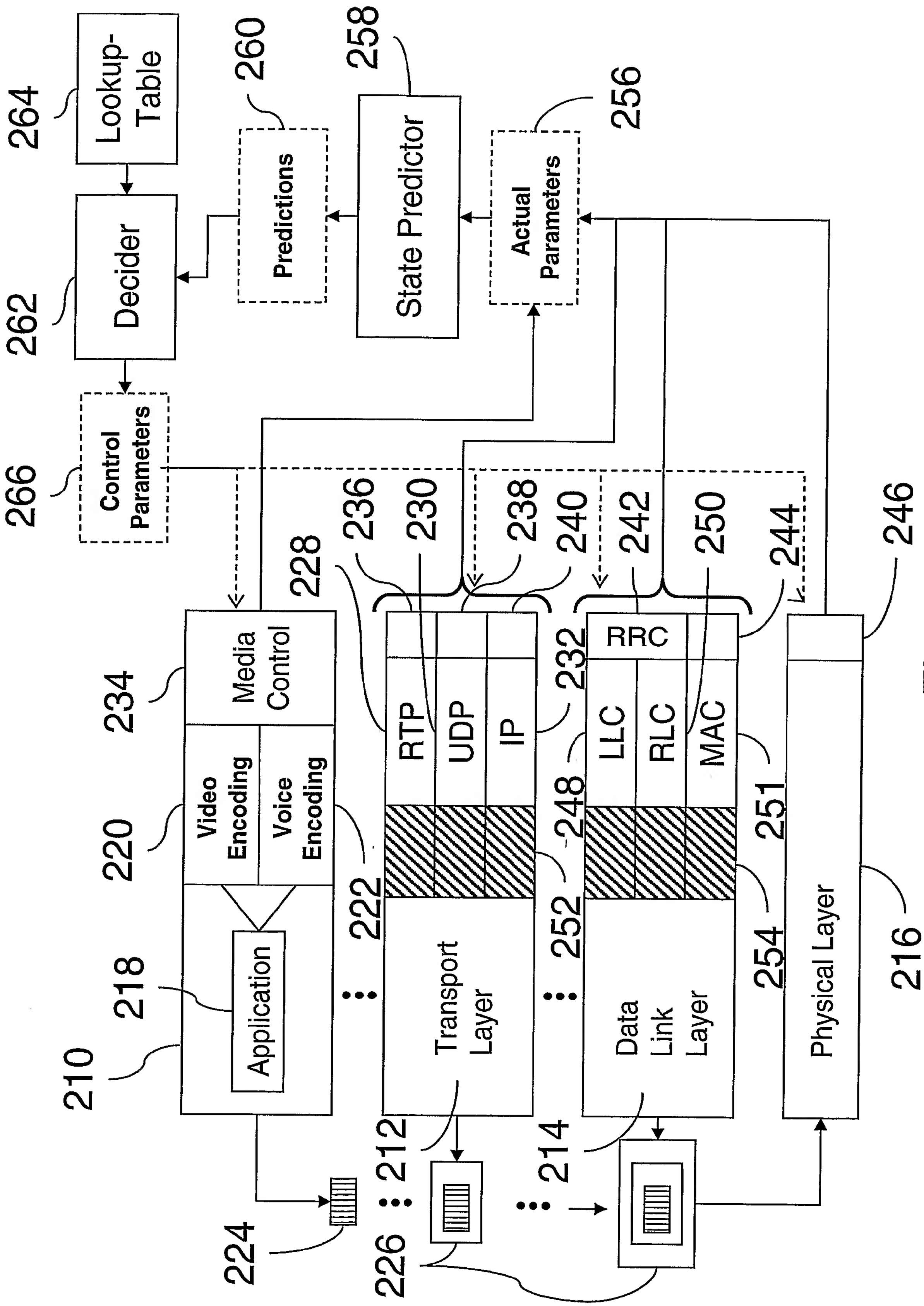

110
Acquisition of data flow parameters
112
Prediction of future data flow
114
Decision on flow optimization
116
Dynamic I-frame forward error correction (FEC)
118
Dynamic I-frame generation
120
Dynamic I-frame generation delaying
122
Dynamic P-frame rate adjusting
124
Dynamic quantization adaptation

Fig. 1


Lookup-Table
Decider
Predictions
State Predictor
Actual Parameters
Control Parameters
Media Control
Video Encoding
Voice Encoding
Application
RTP
UDP
IP
Transport Layer
RRC
LLC
RLC
MAC
Data Link Layer
Physical Layer
264
262
260
258
256
266
234
220
218
210
222
228
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230
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232
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Fig. 2

INTERNATIONAL SEARCH REPORT

PCT/EP2004/008688

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7 H04N7/26 H04N7/24

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 7 H04N

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

EPO-Internal

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	LEI Z ET AL: "A rate adaptation transcoding scheme for real-time video transmission over wireless channels" SIGNAL PROCESSING. IMAGE COMMUNICATION, ELSEVIER SCIENCE PUBLISHERS, AMSTERDAM, NL, vol. 18, no. 8, September 2003 (2003-09), pages 641-658, XP004452903 ISSN: 0923-5965 page 641, left-hand column, paragraph 1 - page 645, right-hand column, paragraph 3.1 page 648, right-hand column, paragraph 4 - page 649, left-hand column, line 18 page 651, right-hand column, paragraph 5.2 page 654, right-hand column, paragraph 6 - page 657, right-hand column, paragraph 7 ----- -/--	1-10

[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

° Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance

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"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
24 February 2005	04/03/2005

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL – 2280 HV Rijswijk Tel. (+31–70) 340–2040, Tx. 31 651 epo nl, Fax: (+31–70) 340–3016	Schoeyer, M

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	WO 02/089516 A (MIRBAHA, RAMIN; MIRBAHA, VAHID) 7 November 2002 (2002-11-07) cited in the application abstract page 5, line 20 - page 6, line 16	1-10
A	YIAO LIU ET AL: "A novel adaptive error control scheme for real time wireless video streaming" INFORMATION TECHNOLOGY: RESEARCH AND EDUCATION, 2003. PROCEEDINGS. ITRE2003. INTERNATIONAL CONFERENCE ON AUG. 11-13, 2003, PISCATAWAY, NJ, USA,IEEE, 11 August 2003 (2003-08-11), pages 460-463, XP010685492 ISBN: 0-7803-7724-9 page 460, right-hand column, paragraph II - page 462, right-hand column, paragraph A	1-10
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INTERNATIONAL SEARCH REPORT

PCT/EP2004/008688

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